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(71) Applicant
STC plc (United Kingdom),
190 Strand, London WC2R 1DU

(72) Inventors
Bjorn Napier Hansen,
Richard William Charlesworth

(74) Agent and/or Address for Service
S. R. Capsey,
STC Patent Department, Edinburgh Way, Harlow,
Essex CM20 2SH

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None

(58) Field of search
H4K

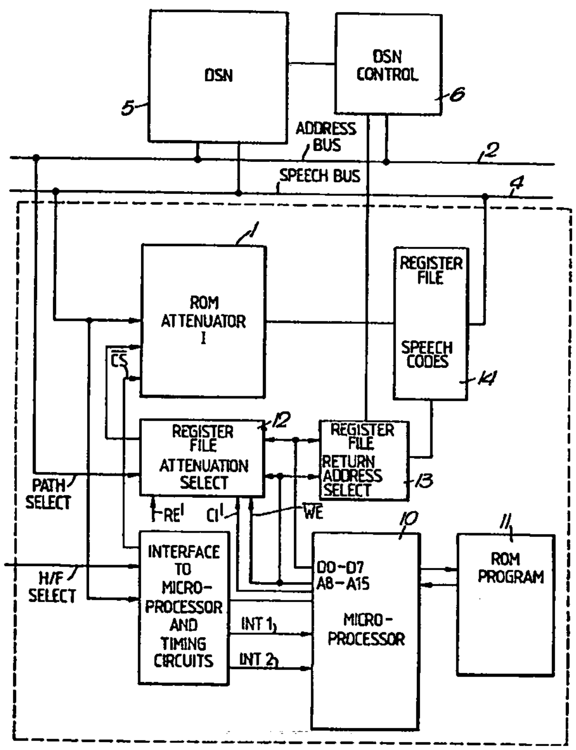
B1

(54) Telephone exchange conferencing

(57) In our earlier Application No. 8305572 (B.N. Hansen 2) and No. 8307143 (B.N. Hansen 1), a handsfree system is described in which the incoming and outgoing speech channels of a handsfree set are monitored. The result of these monitorings assesses which channel has the higher amplitude, and the lower amplitude channel is attenuated, thus avoiding howling. There is also a noise threshold adjustment used in this monitoring.

The above technique is, in the present system, extended to its application to conference calls using handsfree sets. Here the speech amplitudes for all conferees are sampled and the largest is routed to all other conferees, who are attentuated. This technique is also extended to its use where a conference call involves an analogue junction to a remote exchange. As in the single handsfree case, this operation also takes into account the lines' noise condition.

Fig. 1.



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The drawing(s) originally filed was (were) informal and the print here reproduced is taken from a later filed formal copy.

Fig.1.

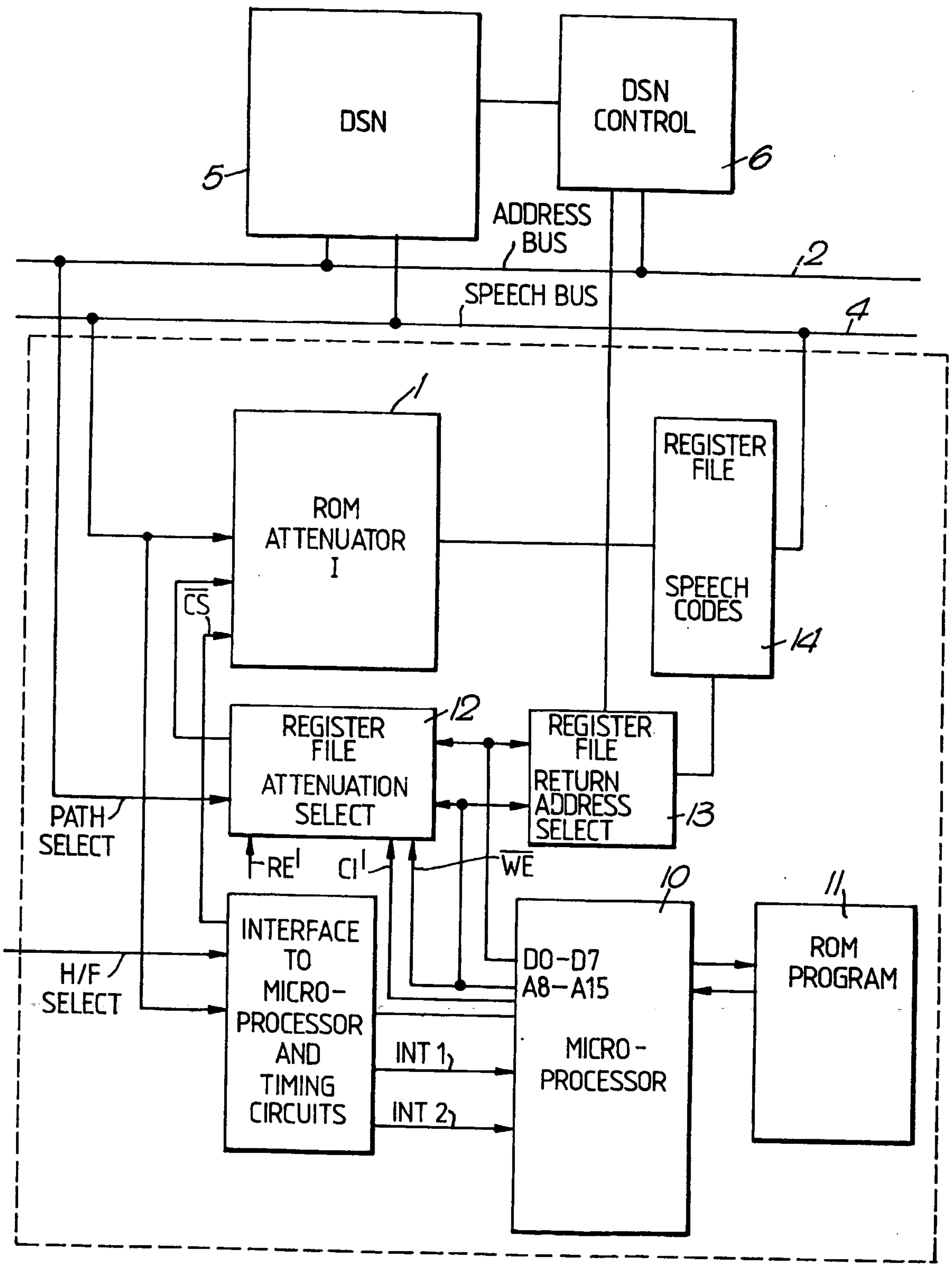
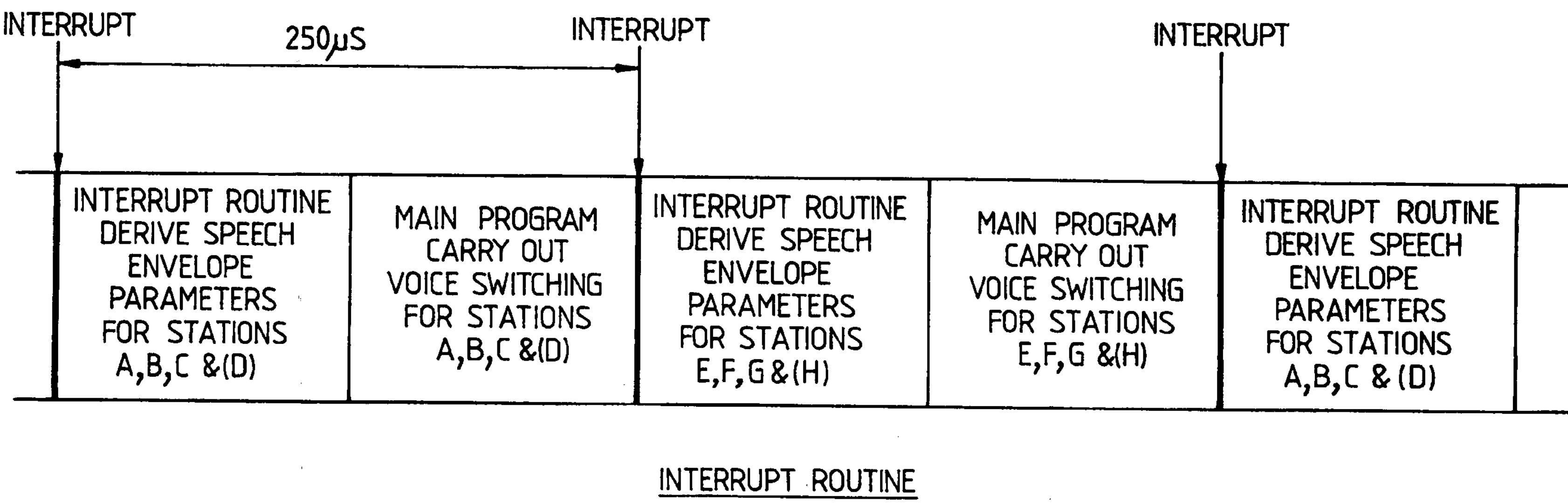

DSN
DSN CONTROL
5
6
ADDRESS BUS
2
SPEECH BUS
4
1
ROM ATTENUATOR I
REGISTER FILE
SPEECH CODES
14
CS
12
REGISTER FILE ATTENUATION SELECT
REGISTER FILE RETURN ADDRESS SELECT
13
PATH SELECT
RE
CI
WE
10
11
INTERFACE TO MICRO-PROCESSOR AND TIMING CIRCUITS
D0-D7
A8-A15
MICRO-PROCESSOR
ROM PROGRAM
H/F SELECT
INT 1
INT 2

Fig. 2.


INTERRUPT
250μS
INTERRUPT
INTERRUPT
INTERRUPT ROUTINE DERIVE SPEECH ENVELOPE PARAMETERS FOR STATIONS A,B,C &(D)
MAIN PROGRAM CARRY OUT VOICE SWITCHING FOR STATIONS A,B,C &(D)
INTERRUPT ROUTINE DERIVE SPEECH ENVELOPE PARAMETERS FOR STATIONS E,F,G &(H)
MAIN PROGRAM CARRY OUT VOICE SWITCHING FOR STATIONS E,F,G &(H)
INTERRUPT ROUTINE DERIVE SPEECH ENVELOPE PARAMETERS FOR STATIONS A,B,C &(D)
INTERRUPT ROUTINE

MAIN PROGRAM

Fig. 3. (part 1).

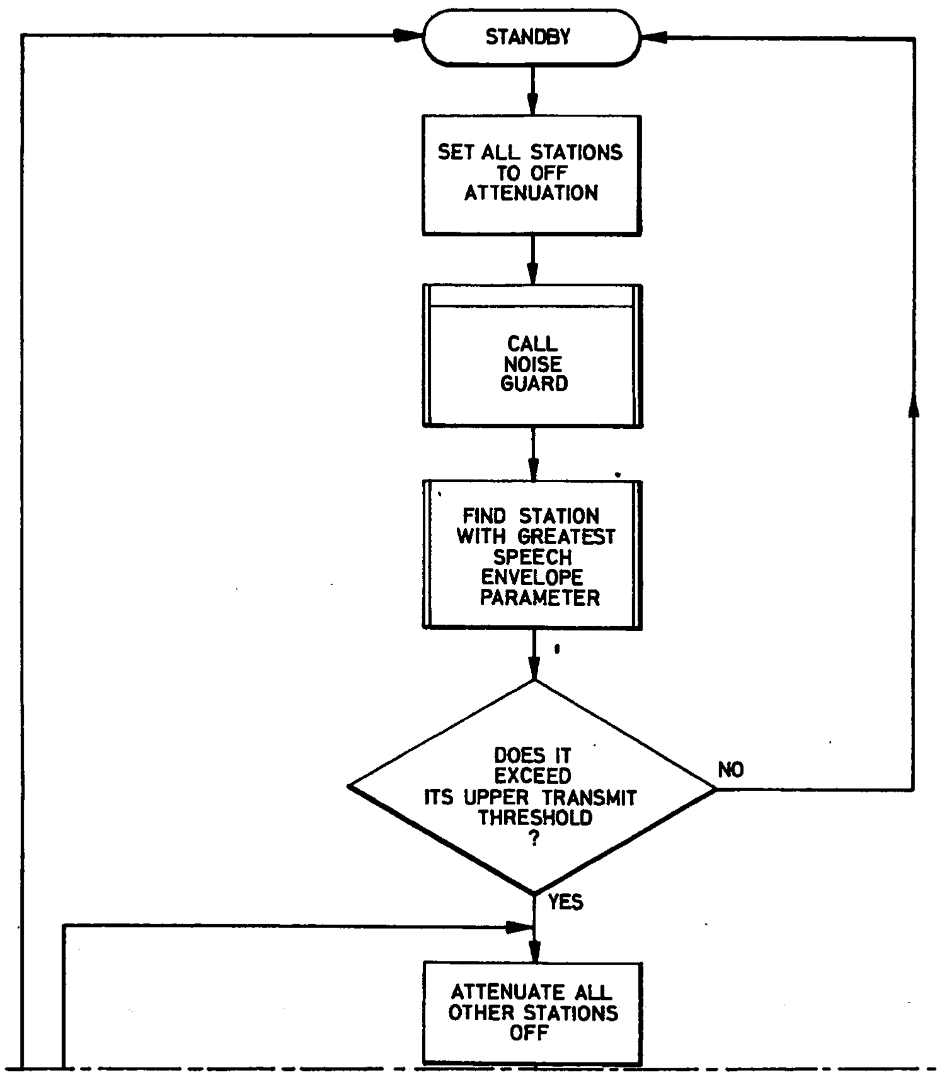


Fig.3.(part 2).

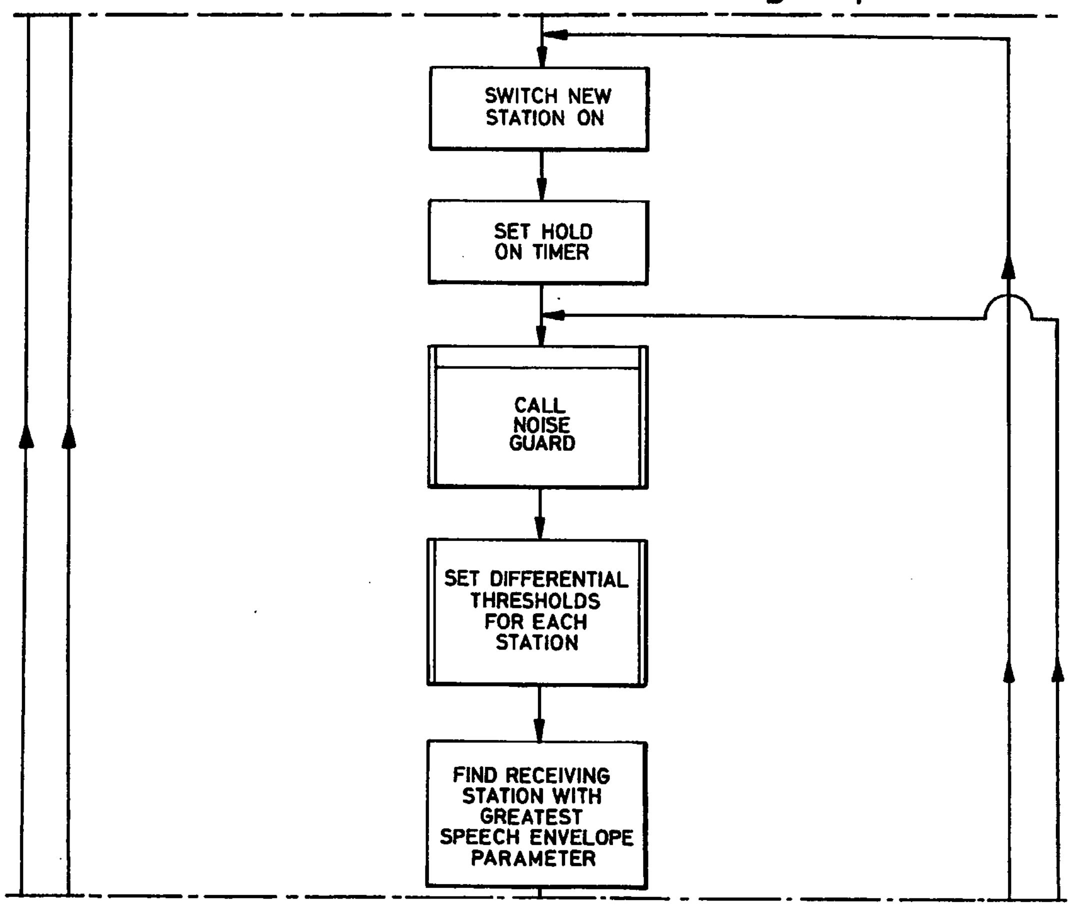
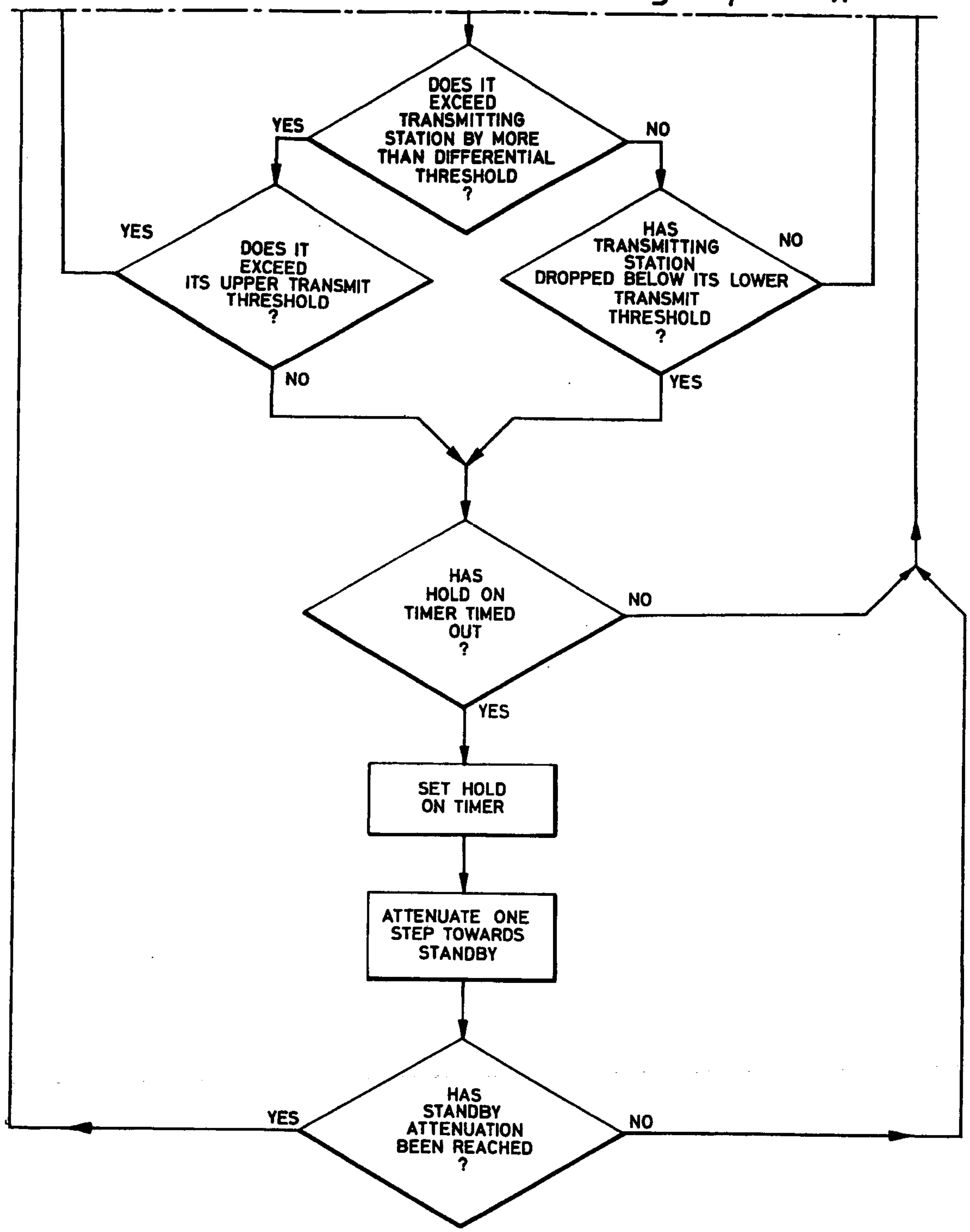
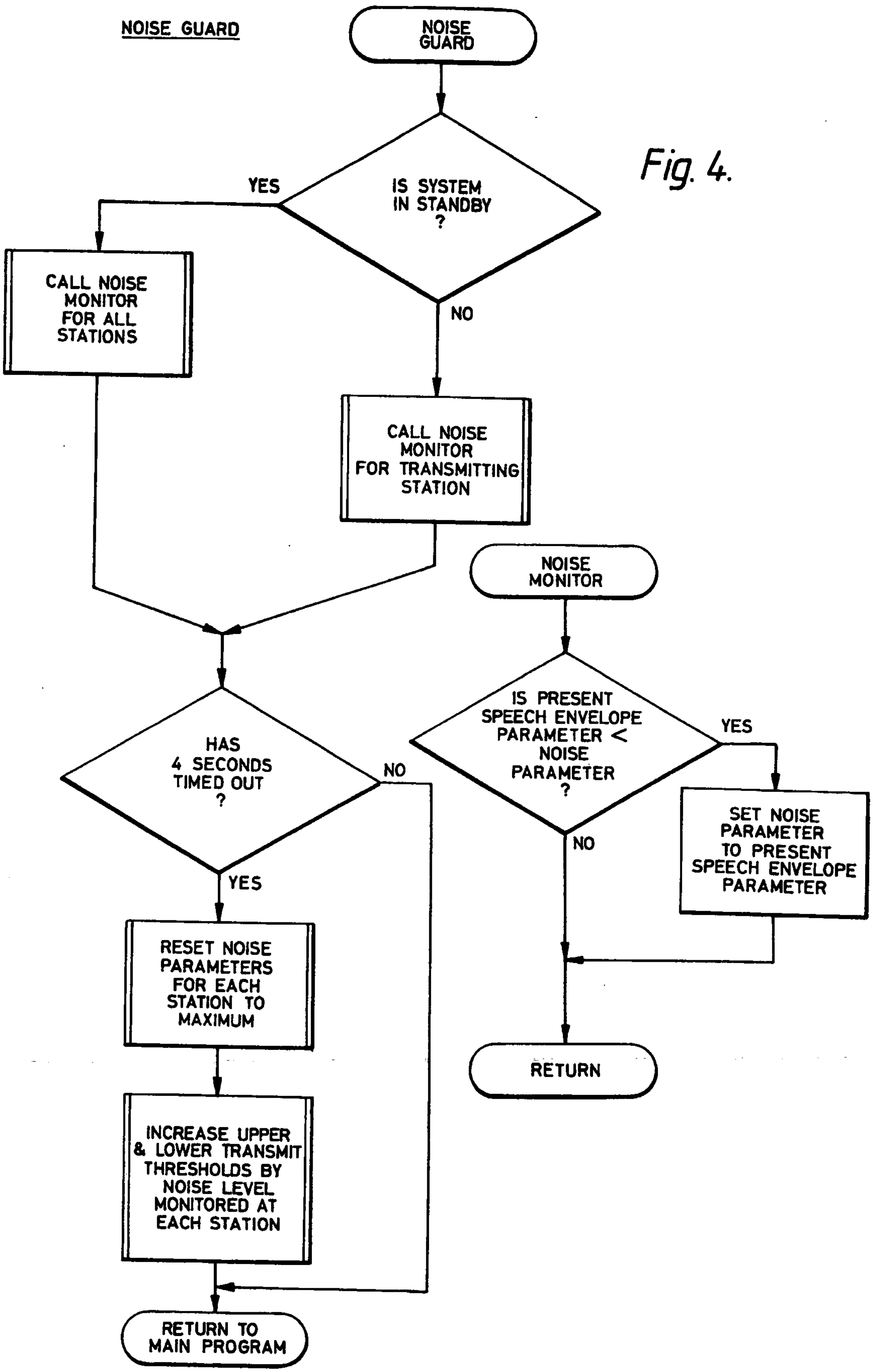

SWITCH NEW STATION ON
SET HOLD ON TIMER
CALL NOISE GUARD
SET DIFFERENTIAL THRESHOLDS FOR EACH STATION
FIND RECEIVING STATION WITH GREATEST SPEECH ENVELOPE PARAMETER

Fig. 3. (part 3).


DOES IT EXCEED TRANSMITTING STATION BY MORE THAN DIFFERENTIAL THRESHOLD ?
YES
NO
DOES IT EXCEED ITS UPPER TRANSMIT THRESHOLD ?
YES
NO
HAS TRANSMITTING STATION DROPPED BELOW ITS LOWER TRANSMIT THRESHOLD ?
NO
YES
HAS HOLD ON TIMER TIMED OUT ?
NO
YES
SET HOLD ON TIMER
ATTENUATE ONE STEP TOWARDS STANDBY
HAS STANDBY ATTENUATION BEEN REACHED ?
YES
NO



NOISE GUARD
NOISE GUARD
IS SYSTEM IN STANDBY ?
YES
NO
CALL NOISE MONITOR FOR ALL STATIONS
CALL NOISE MONITOR FOR TRANSMITTING STATION
HAS 4 SECONDS TIMED OUT ?
NO
YES
RESET NOISE PARAMETERS FOR EACH STATION TO MAXIMUM
INCREASE UPPER & LOWER TRANSMIT THRESHOLDS BY NOISE LEVEL MONITORED AT EACH STATION
RETURN TO MAIN PROGRAM
NOISE MONITOR
IS PRESENT SPEECH ENVELOPE PARAMETER < NOISE PARAMETER ?
YES
NO
SET NOISE PARAMETER TO PRESENT SPEECH ENVELOPE PARAMETER
RETURN

Fig. 4.

Fig. 5.

INTERRUPT PROGRAM

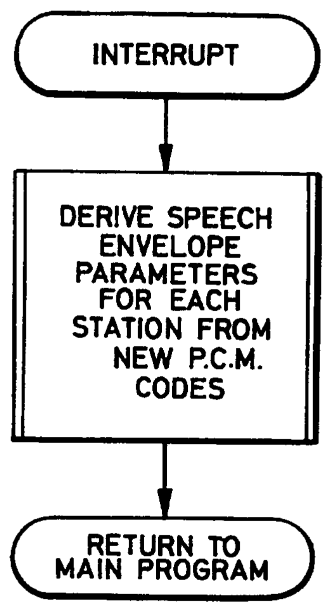
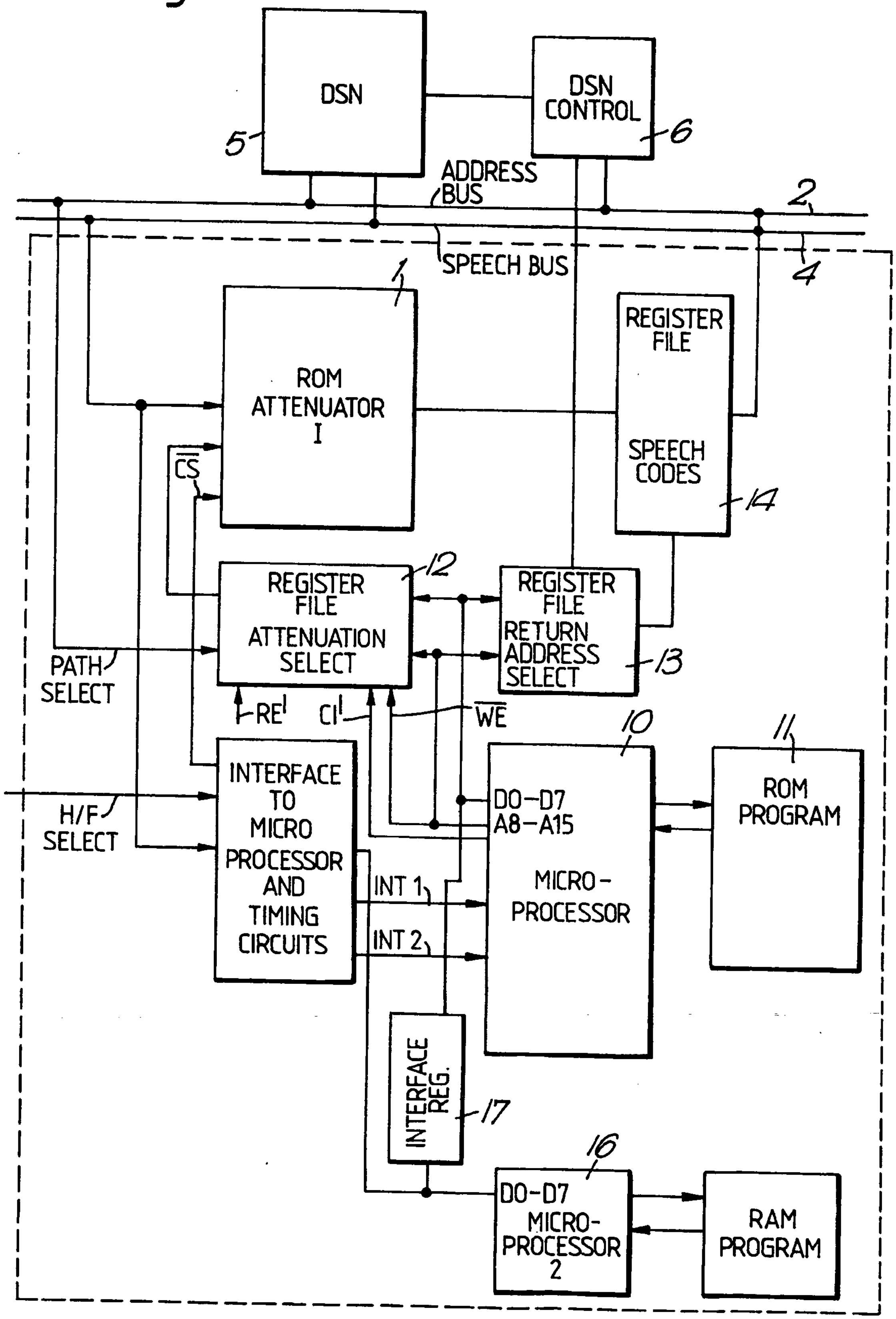


Fig. 6.



DSN
5
DSN CONTROL
6
ADDRESS BUS
2
SPEECH BUS
4
ROM ATTENUATOR I
1
REGISTER FILE
SPEECH CODES
14
CS
REGISTER FILE ATTENUATION SELECT
12
REGISTER FILE RETURN ADDRESS SELECT
13
PATH SELECT
RE
CI
WE
10
11
INTERFACE TO MICRO PROCESSOR AND TIMING CIRCUITS
H/F SELECT
D0–D7
A8–A15
MICRO-PROCESSOR
ROM PROGRAM
INT 1
INT 2
INTERFACE REG.
17
16
D0–D7
MICRO-PROCESSOR 2
RAM PROGRAM

SPECIFICATION

Telephone exchange conferencing

This invention relates to a conference circuit arrangement which enables more than two digital telephones to be interconnected through a common switching network, especially when the network is a digital one, and the telephones are of the handsfree type.

Existing conference circuits applied to networks in which speech is transmitted digitally, such as digital PABX's and key systems, work well when conference calls are set up between handset telephones. However, when a handsfree telephone or a connection to an analogue trunk to another exchange via a four wire to two wire interface is involved, reflected speech signals from loudspeaker to the microphone on the handset and reflected signals across the interface cause instability and hence oscillations and speech distortion. Speech distortion is especially noticeable noticeable in the presence of background noise.

These problems may be resolved by attenuating the transmit speech paths of all listening stations when one party is talking, thus significantly reducing or eliminating the disturbing reflection. Arrangements for doing this are described in our Application No. 8305572 (B.N. Hansen 2) and No. 8307143 (B.N. Hansen 1).

An object of the invention is to extend the basic principles of these two applications to conferencing.

According to the invention there is provided a conference circuit arrangement for use in an automatic telephone exchange, in which the speech signals in the speech paths of the subscribers engaged in a conference call are individually sampled and the resulting samples are compared with preset speech thresholds, and in which the subscriber station with the greatest speech amplitude which at the same time exceeds its transmit threshold, which threshold is defined by the current noise condition of the station, causes the transmitted speech signal from that station to be switched to the other stations involved in the call while at the same time the transmitted signals from those other stations are attenuated.

Thus in the arrangement to be described below, control of the speech signal attenuation is based on the data processor controlled voice switching techniques of the above-mentioned Applications. Briefly, speech envelope parameters are derived from each of the participating stations, and control decisions are made on the basis of comparisons between the individual speech envelopes and preset speech thresholds. The station with the greatest speech envelope parameter which at the same time exceeds its transmit threshold causes the control software to switch the transmitted speech codes to the other participating stations while attenuating transmit signals from those other stations.

A practical realisation of this method is described herein for a three or four party conference. The actual circuit allows up to eight such conferences to occur using digital telephones, which may be in either handset mode or handsfree mode, and a conference may include access to an analogue exchange line. The application areas are digital key systems, digital PABX's, etc. where a number of digital telephones are interconnected via a common digital switching network (DSN). The conference call control circuit is centrally located and is used as a shared facility accessible to a number of telephones.

An eight channel, i.e. sixteen station version has been built with a speech sampling proportion of one in eight. In addition the software has been modified so that one of the two programs mentioned provides for the single handsfree conferencing. This allows the same circuit to provide a mix of single handsfree conversations and conference calls, where a conference call occupies two channels, e.g. in the eight channel case one conference and six single handsfree conversations or two handsfree conferences and four single handsfree conversations etc. The software automatically adjusts itself according to the required conditions by control codes transmitted from the DSN, which again receives appropriate signalling codes from the individual digital sations when calls are being set up.

The programs are time shared between the single handsfree channels or conferences.

The invention will be described in its application to a system such as that our our Application No. 8305572 (B.N. Hansen 2), and No. 8307143 (B.N. Hansen 1), so brief description of the arrangements of these applications follow. However, it must be appreciated that the present invention is usable in other systems.

In the loudspeaking telephone arrangement of our Application No. 8305572, there are separate channels for outgoing and incoming speech. During a telephone conversation these channels are sampled at intervals, e.g. one per cycle of the TDM frame. The results of these samplings, which represent the speech amplitudes in these channels and are in digital form, are each compared with a preset threshold. The background noise level is also sampled by monitoring the channels when they are not conveying speech, and the results of these comparisons, which are also in digital form, are used to adjust the speech channel thresholds. Each channel includes an attenuator which is so adjusted that attenuation is reduced in the presence of speech and increased in the absence thereof. The adjustment of these thresholds on the basis of the background noise enables the current states of the channels to be taken into account.

The arrangement of our Application No. 8307143 follows the principles of Application No. 8305572, above, but exploits the fact that it is not necessary to monitor *all* of the speech signals to derive adequate information about the states of the two speech channels. Hence only one speech sample in four is used, although it is appreciated that a different proportion could be used. This enables one microprocessor to serve two "handsfree" calls al-

ternately. Although this increases the amount of memory used, this costs less than the microprocessor saved by the use of this technique.

The conference circuit arrangement to be described herein uses as its central control circuit an arrangement, shown in Figure 1 of the accompanying drawings, which is similar to the central control arrangement of the above-mentioned Application No. 8307143, but with an added block 13 and the connections thereto. Figures 2 to 5 are software flow sheets, useful in explaining the operation. Figure 6 is another embodiment of the invention.

Two handsfree control programs are run alternately by the same microprocessor, allowing two simultaneous handsfree conversations to take place. This halves the number of microprocessors needed at the cost of doubling the software memory requirements, but the additional memory needed costs much less than an additional microprocessor.

Software control of channel gains is done by continuously monitoring the speech samples from both directions and deriving from these monitorings parameters representing the speech envelopes. The envelope parameters are, as in application No. 8305572, compared with set thresholds and the appropriate speech channel is enabled or partially disabled by controlling the attenuation of the speech paths. The derivation of the parameter representative of the speech envelope does not need every speech sample to be monitored, but as described, every fourth sample gives sufficient information as to the current speech patterns in the channels. Hence sufficient processing time is available to allow adequate gain control of two handsfree channels simultaneously.

The two-channel digital handsfree facility for one extension includes an attenuator for digital signals formed by a Read Only Memory 1. The input code from the address bus 2 of the system provides the ROM address and the corresponding output is a code representing an attenuated version of the input. There is also a control circuit which includes the microprocessor 10 and memory, connected to the speech bus 4 and address bus 2 of the telephone system. This system has a digital switching network 5 and its control unit 6. During a handsfree conversation, digital codes representing speech amplitudes from caller A to called party B and vice versa are continuously fetched from memory locations in the network addressed through the attenuator 1 and written back into other memory locations in the network. Initially this occurs without interference from the microprocessor and each channel is attenuated by a nominal amount, i.e. the channel is in standby. In the meantime the control circuit monitors every fourth transmit and receive sample from each channel and from this information it derives control parameters which set the amount of attenuation of each sample as it is addressed through the ROM attenuator 1. All further settings of the channel gains are now determined by the handsfree controller circuit according to algorithms placed in the controlling software programs.

In Figure 1, microprocessor 10 and the ROM program memory 11 are shown as separate components, but the method is applicable when using a single-chip microcomputer or any other suitable data processing device under software control. The software consists basically of two identical, but independent, main programs each containing one sub-routine or interrupt routine. The speech path gain settings are determined by algorithms in the main programs. The interrupt routines monitor the current speech patterns in the two channels and pass this information to the main programs for further processing. The sampling rate of the speech channels is assumed to be I/T, and the program flow is interrupted very 2T sec., alternating between the two main programs by timing pulses from the interface circuit which are applied to two interrupt inputs INT1 and INT2 on the processor. The interrupt routines run at the beginning of each 2T sec. processing period. During execution of the interrupt routine, signal amplitude information from the two directions of one speech channel is converted to an approximation of the speech envelope whose parameters are passed to the main program. Processing then continues in the first main program until expiry of the 2 T sec. period, during which time the speech envelope parameters are compared with set thresholds and the appropriate channel is enabled or partially disabled by information passed to the attenuation select registers 12. Processing of the first main program is interrupted at the end of the 2 T sec. period by a timing pulse on input INT2, after which interrupt routine 2 starts and monitors speech information in the two directions on the other channel. As before, this information is passed to the other main program for processing. This other main program continues execution at the address following the address at which it was interrupted by a timing pulse on INTI. Similarly for the first main program.

The results of the processings in the two main programs are four parameters, two from each program, which are related to the attenuation levels needed for the two speech directions in each channel. These are stored in a temporary register unit, the attenuation select register 12, Figure 2. The memory are of the ROM attenuator 1 is divided into a number of attenuation level areas. Within each such area the nominal speech sample digital code has been converted to a corresponding code with the appropriate attenuation, e.g. the (A-law) PCM sample code 11001010 when subjected to 6 dB attenuation becomes 11011010. These new codes replace the incoming codes to the ROM attenuator 1, the output of which is a sequence of codes representing a speech pattern identical to the incoming speech pattern, but reduced in amplitude.

The ROM attenuator 1 operates as follows. Incoming speech sample codes form part of the address of the replacement code. The remaining part of the address comes from the register 12, which enables access to specific areas of the ROM attenuator 1. Correct correspondence between speech sample and associated attenuation code is

ensured by the sample's address code supplied by the DSN 5 and input as an address to the register 12. The attenuated output code is latched and then written back into its appropriate register in the DSN 5 by the DSN control unit 6.

As already indicated, the proportion of speech samples monitored may be other than one in four. Thus for delta modulation type system, where the sampling rate is relatively high, it may be possible for the proportion of speech samples monitored to be reduced.

The hardware needed to implement the conferencing facility, in addition to that of our Application No. 8301743, is a Return Address Select file 13, which is a RAM which is continually updated with the address of the current transmitting station as the conversation proceeds.

During one of the receive station time slots, its address is presented to the handsfree control circuit from the DSN control unit 6. This address is input to the return address select file 13, which outputs the address of the transmitting station. This in turn selects the speech code of the transmitting station from the speech code file in the block 14, which replaces the latch block in the system of Application No. 8301743. This speech code is then conveyed to the receive station under control of the DSN 5.

During the transmit time slot, an attenuated code from one of the receive stations is selected in a similar way and conveyed to the transmitting station.

The software algorithms which continuously monitor the speech codes and from these determine the return address are described below. Note that only one program in the block 11 is used for conferencing, as compared with two in the handsfree case.

The digital handsfree conference software consists of three main parts :

(1) *Interrupt routine* - which derives an approximation to the speech envelope of each party or station.

(2) *Main program* - which carries out voice switching according to the speech envelopes derived in the interrupt routine.

(3) *Noise guard* - which is a subroutine from the main program, which measures the background noise at each station and adjusts the switching threshold used in the main program accordingly.

General note

The interrupt routine is serviced every time a new set of PCM codes from the stations in conference are input to the conference circuit. Assuming every PCM sample is monitored, this will be every 125 uS (8 kHz sampling frequency). This interval may be increased if say every fourth or every eight PCM sample is monitored, as in the handsfree system just described. Thus the system can be expanded to cater for multiples of 3-or 4-party conferences. Figure 2 shows the general interrupt timing and channel time allocation for a system which caters for two, three or four party conferences, monitoring every fourth PCM sample of each station. The interrupt is serviced every 250 uS. The processor changes from processing one conference channel to the other at this point. Thus for a given conference channel, the PCM samples are monitored every 500 uS (every fourth sample).

For a particular channel the conference can be in one of three states at any one time : (1) Standby, where all stations are equally attenuated; (2) One particular station transmitting, the others receiving; and (3) Slow ramping to standby, a transitional state between (2) and (1). When no one is talking the system is in standby. Each station has an upper and lower transmit threshold. To switch into transmit state a station's speech level must exceed its upper threshold. When the speech level drops below the lower threshold the system ramps back to standby. During the transmit state, a test is included which allows any of the receiving stations to break in on the transmitting station.

Main program (Figure 3)

The Main Program first enters a standby loop in which all parties (stations) are equally attenuated. During this loop the Noise Guard is called, the operation of which is described later. During the standby loop the party with the greatest speech envelope parameter (derived in the interrupt routine) is compared with its upper transmit threshold. If it does not exceed this, the loop is repeated. If it exceeds its upper threshold the other parties are attenuated off, and it is switched on to its active attenuation. This is done by sending the relevant attenuation codes so the Attenuation select file and also by sending to the Return Address select file the addresses from which to fetch the receive PCM codes.

The program now enters a differential comparison loop in which again the noise guard is called. This loop allows receiving stations to break in on the transmitting station. Each party has a differential threshold set according to whether that party is in handsfree mode, handset mode or is a trunk connection, and in the case of handsfree, according to the volume control. The receiving parties' speech envelope parameters are monitored, the greatest being taken and compared with that of the transmitting station. For a receiving station to break in, its speech envelope parameter must exceed that of the transmitting station by its differential threshold. It must also exceed its upper transmit threshold. If no break-in occurs the transmitting station is tested against its lower transmit threshold. If it falls below this threshold, a 128 mS hold in active state is initiated after which the station is slow ramped to standby attenuation, whereupon the program re-enters the standby loop. During ramping the transmitting station may regain access if it exceeds its lower transmit threshold.

Noise Guard (Figure 4)

The Noise Guard looks for the minimum speech envelope parameter for each station over a four second period. The minimum for a given station is

only looked for when that station is transmitting, or when the system is in standby. This is down to avoid problems of background noise at one station being transmitted to another and being picked up by the microphone of the receiving station, thus monitoring an artifically high noise level.

The minimum speech envelope parameter for a given station is taken as the background noise level at that station. This level is used to adjust the upper and lower transmit thresholds so that the lower threshold lies just above the background noise level, enabling the station to drop back to standby.

Interrupt Routine (Figure 5)

The interrupt routine is serviced every time a new PCM code for each station is input to the conference circuit. The routine uses these codes to derive an approximation to the speech envelope of each station. These speech envelope parameters are then used in the main program to control the voice switching of the conference.

Eight channel handsfree/handsfree conferencing control circuit

An eight channel version is shown in Figure 6, in which, because of the additional processing time required to calculate the speech envelope parameters of the individual speech paths for speech paths, an additional microcomputer 16 is used.

Here the speech envelope processing carried out by the interrupt routine in the two channel version is replaced by the separate microcomputer, which is a slave working in parallel with the main program processor 10. The slave processor inputs sampled speech codes, calculates speech envelope parameters and transfers these at regular intervals via an interface register 17 to the master processor 10, where the speech path attenuation control decisions are made.

This method significantly expands the number of channels that the circuit can cope with at the cost of an additional and relatively low priced microcomputer.

The software algorithms used are briefly described below.

The processing for the eight channel system is split between the two microprocessors. The second microprocessor 16 is a slave processor whose sole task is to derive speech envelope parameters for each station, sixteen in all. These are then passed sequentially to the interface register 17 from which the master processor 10 reads and carries out the voice switching. The two microprocessors are interrupt driven with the timing chosen so that the interface register 17 is never written to and read from at the same time.

The timing used is such that the two processors are processing one channel behind the other. Thus while the slave processor is deriving speech envelopes for channel O for one pair of stations, the master processor 10 is still carrying out voice switching for channel 7 for the previous pair of stations. When the speech envelope parameters are written to the interface register 17 the channel number to which they refer is also sent to ensure that the two processors keep in step.

The master processor 10 has two main programs in additional to its interrupt routine. One deals with the switching of an ordinary single handsfree connection (one channel of the eight), the other deals with a conference connection. The program has a routine which decides which two channels are in a particular conference connection and carries out voice switching accordingly as in the flow chart in Figure 3. A conference connection occupies two channels.

The interrupt routine for the slave processor decides at the end which main program to return to, depending on the type of connection for that channel.

CLAIMS

1. A conference circuit arrangement for use in an automatic telephone exchange, in which the speech signals in the speech paths of the subscribers engaged in a conference call are individually sampled and the resulting samples are compared with preset speech thresholds, and in which the subscriber station with the greatest speech amplitude which at the same time exceeds its transmit threshold, which threshold is defined by the current noise condition of the station, causes the transmitted speech signal from that station to be switched to the other stations involved in the call while at the same time the transmitted signals from those other stations are attenuated.

2. A conference circuit arrangement for use in a digital automatic telecommunication exchange, in which some at least of the lines served by the exchange are of the handsfree type, in which for each said handsfree line separate paths are provided for incoming and outgoing speech, in which when a call is in existence which involves a said handsfree line the two speech paths for that line are monitored to assess which one has the higher amplitude, in which the result of said assessment causes the path with the lower amplitude to be attenuated as compared with the path with the higher amplitude, in which the speech in the speech paths of lines involved in a conference call is individually sampled, in which the line with the greatest speech amplitude which at the same time exceeds that line's transmit threshold, which threshold is defined by the current noise condition of the station connected to that line, causes the transmitted speech signal from that line to be switched to the other lines involved in the call while at the same time the transmitted signals from those other stations are attenuated, and in which the monitoring and associated control operations for conference purposes occur in a similar manner to those used for handsfree operation.

3. A circuit arrangement as claimed in claim 1 or 2, and in which one or more of the lines involved in a conference call can be an analogue line, such as an analogue line to a remote exchange.

4. A conference circuit arrangement for use in

an automatic telephone exchange, substantially as described with reference to the accompanying drawings.

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